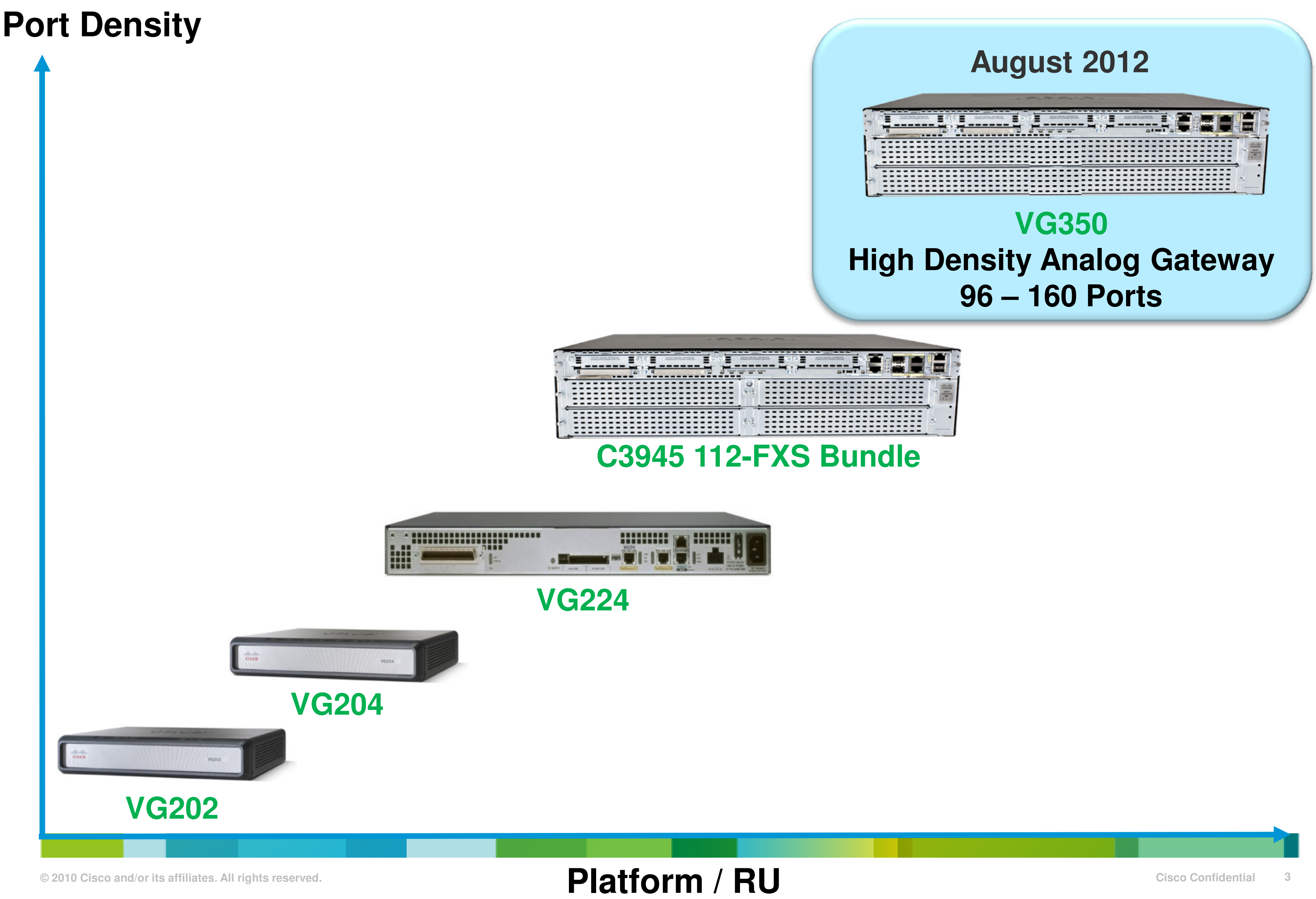
SRTG Update

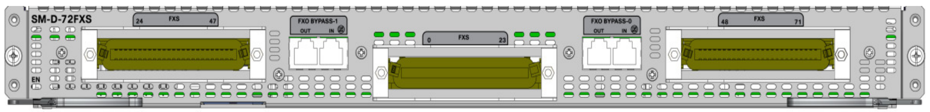
Cisco VG350

High Density Analog Voice Gateway

Cisco Analog Voice Gateway Portfolio



High Density Analog Gateway Solution



SM-D-72FXS
72 Port FXS Double Wide Service Module

VG350
High Density Analog Gateway

SM-D-48FXS-E
48 Port FXS OPX-Lite Double Wide Service Module

Configuration	VG350		
	SM 1	SM 2	Total num of ports
1	SM-D-72FXS	SM-D-72FXS	144
2	SM-D-72FXS	SM-D-48FXS-E	120
3	SM-D-48FXS-E	SM-D-48FXS-E	96

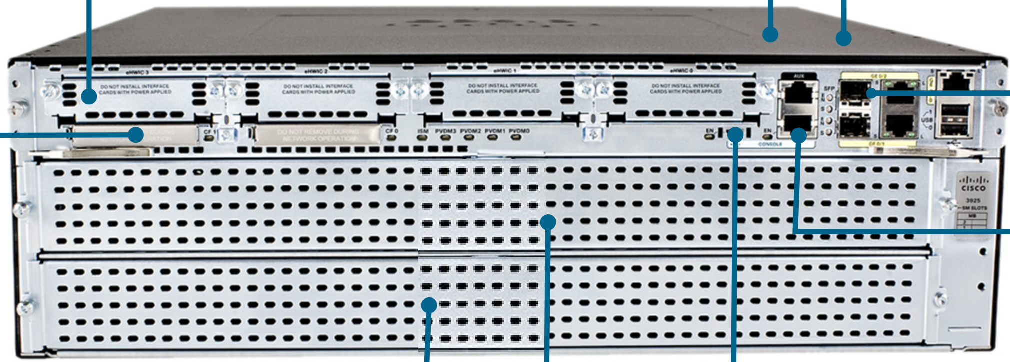
HWIC slots will support FXS, FXS-E and FXO interfaces

Cisco Voice Gateway 350

4 EHWIC Slots
- Support additional FXS and FXO VICs

PVDM3 DSP Modules
- Latest and greatest DSP technology
- Noise Cancelation
- Acoustic Shock Prevention

Redundant Power Supply Option



3 GE Ports

2 Flash storage slots

Standard console and aux ports

2 Double Wide Service Module Slots
- High Density Analog DWSM Slots
- Support
  - 72 Port FXS DWSM
  - 48 Port OPX-Lite FXS DWSM

USB Port
- Console over USB
- Convenient Storage

Cisco VG350 Highlights

- Modified ISR 3945 chassis dedicated for Voice Gateway function
- Software feature parity with VG2xx
  - Voice, fax and modem
  - Comprehensive set of codecs
  - SIP, H.323, SCCP and MGCP protocols
  - TLS / SRTP
  - Supported with CUCM 7.1.5, 8.0.3, 8.5.1, 8.6.2 and 9.0.1
  - Supported with CUCME / SRST 7.1, 8.0, 8.1, 8.5, 8.6, 8.8, 9.0 and 9.1
- HWIC slots can be used for additional FXO and FXS ports
  - VIC2-xFXO, VIC3-xFXS/DID and VIC3-2FXS-E/DID (Long loop length)
- Supports Online Insertion and Removal (OIR) of DWSMs
- Redundant Power Supply option
- Lightening protection
- IOS Release: 15.2(4)M [PI19]

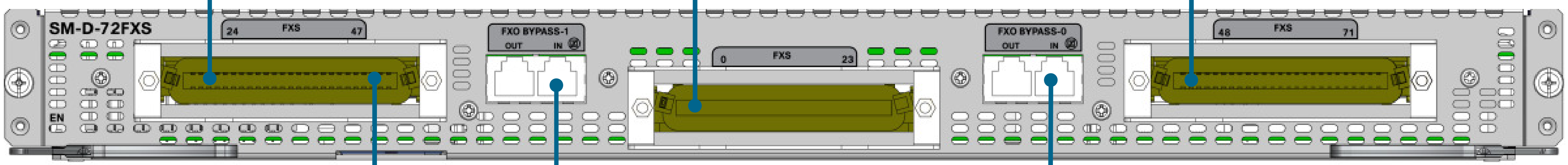
72 Port FXS DWSM

SM-D-72FXS

Ports 24 – 47
- **Ports 46 & 47 connected to PSTN FXO Bypass failover ports**

Ports 0 – 23
- **Ports 0 – 3 support long loop length**

Ports 48 – 71



Standard RJ21 Connectors

2 PSTN FXO Bypass Failover Ports
- **Port 46 ←→ FXO Bypass Port 0**
- **Port 47 ←→ FXO Bypass Port 1**
- **OUT ←→ FXO Port**
- **IN ←→ PSTN**

- SM-D-72-FXS

  72 port FXS Double Wide Service Module

  Loop length – 3,300 ft with 26 AWG

  First 4 ports will support OPX-Lite long loop length

    11,000 ft with 26 AWG

    18,000 ft with 24 AWG

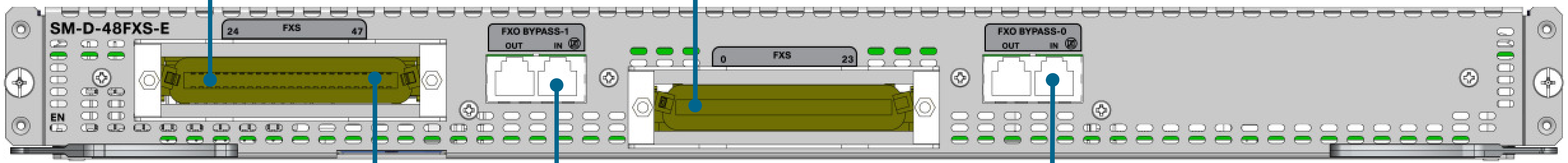
48 Port OPX-Lite FXS DWSM

SM-D-48FXS-E

Ports 24 – 47

- **Ports 46 & 47 connected to PSTN FXO Bypass failover ports**

Ports 0 – 23



Standard RJ21 Connectors

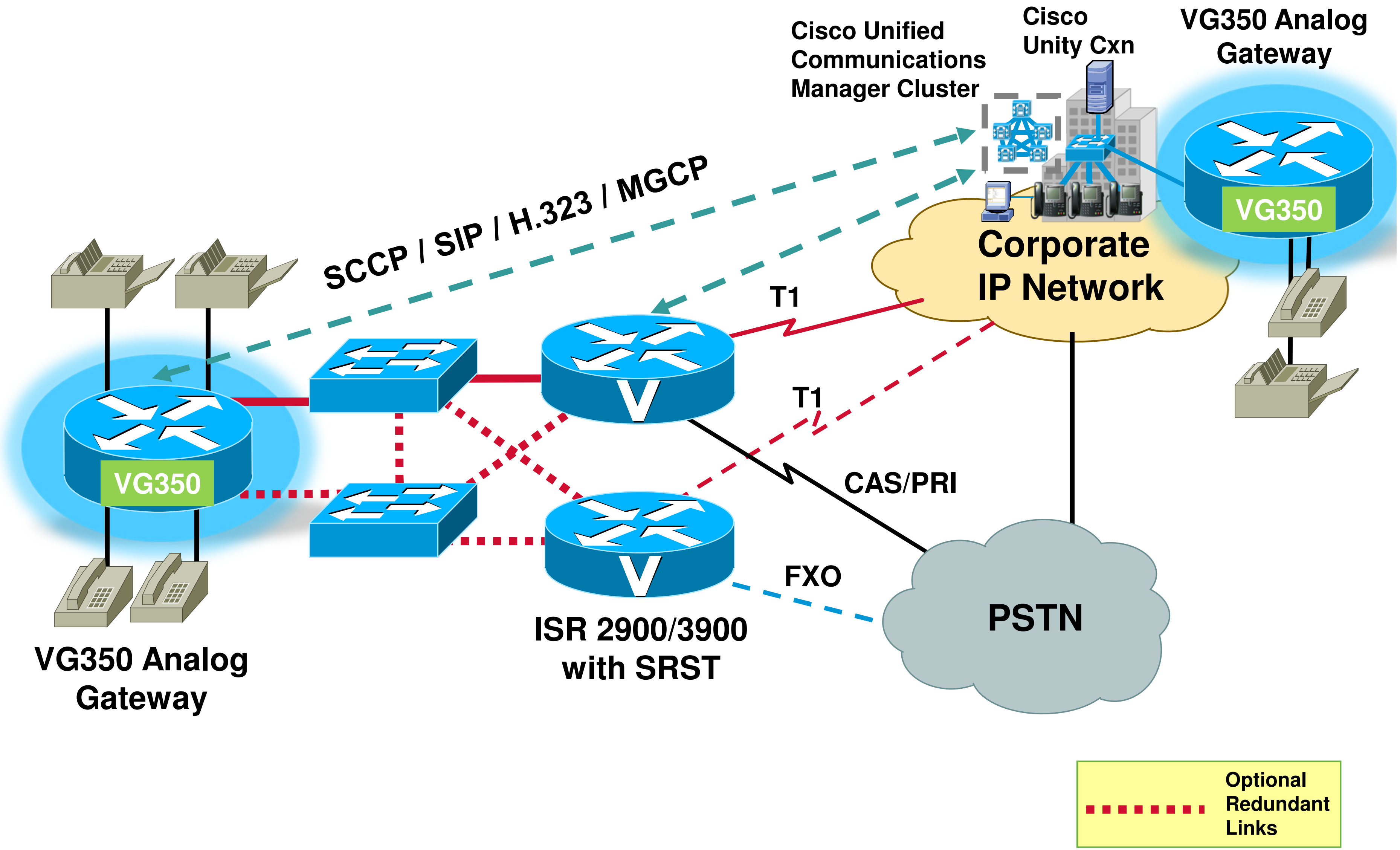
2 PSTN FXO Bypass Failover Ports

- **Port 46 ←→ FXO Bypass Port 0**
- **Port 47 ←→ FXO Bypass Port 1**
- **OUT ←→ FXO Port**
- **IN ←→ PSTN**

- SM-D-48-FXS-E
  - 48 port OPX-Lite FXS Double Wide Service Module
  - All ports will support long loop length
    - 11,000ft with 26 AWG
    - 18,000ft with 24 AWG

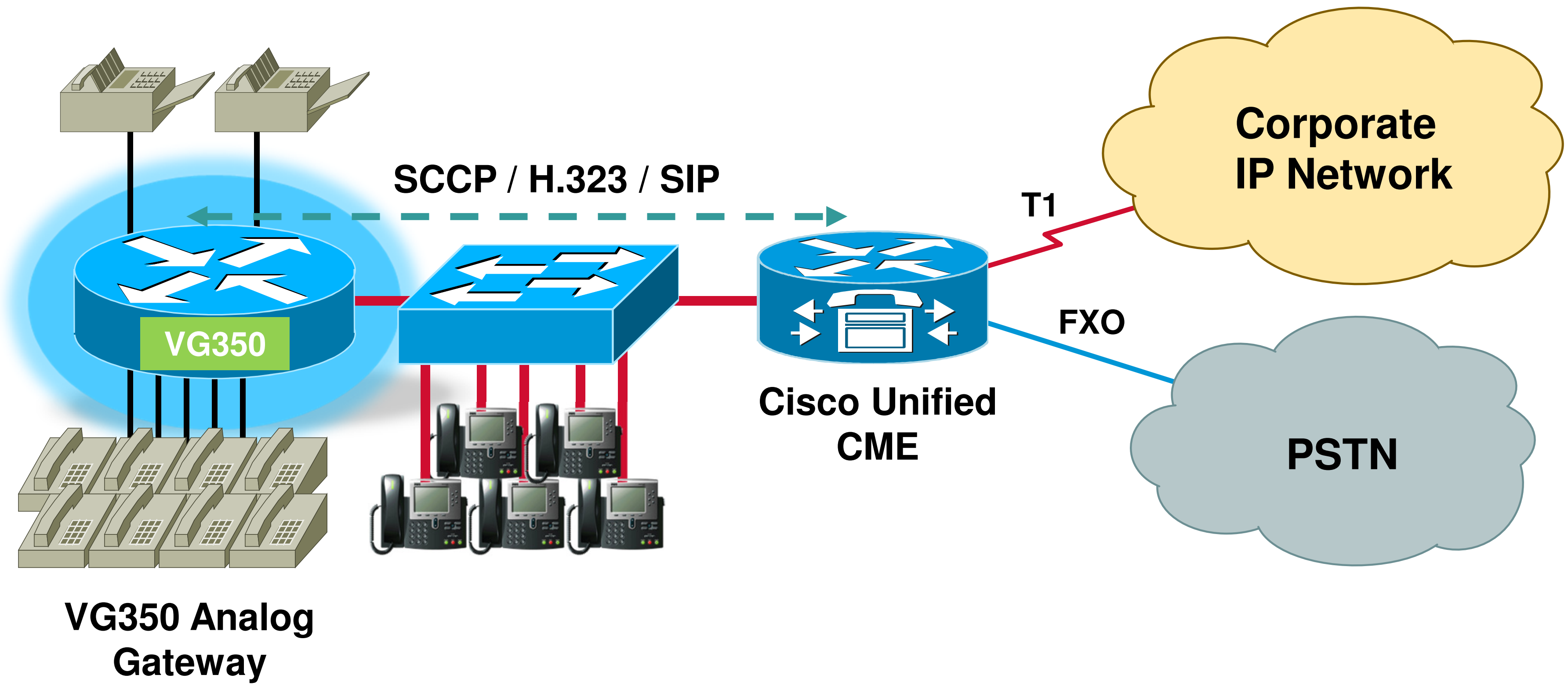
VG350 Deployment Topology

Cisco Unified Communications Manager



VG350 Deployment Topology

Cisco Unified Communications Manager Express

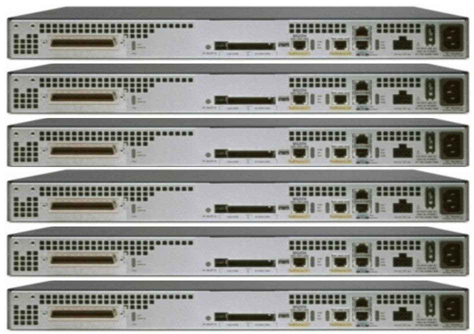


Energywise Support

- Supported at DWSM, EHWIC and PVDM3 module levels
- Modules can be powered off at scheduled times to conserve energy
- During “power off” period, FXO Bypass Failover will still be able to provide dial tone on ports 46 and 47
- CLI is consistent with ISR G2

High Density Benefits

50%
savings in Rack Space



6x VG224 (6 RU)

1x VG350 (3 RU)

83%
savings in Infrastructure & Management

- 6 to 1 savings in
  - Switch port
  - Power supply
  - Console connection
  - Configuration change
  - Software update

Deploying & Managing

- Plug-n-play
  - Learn CUCM IP address from DHCP response
  - Download configuration from CUCM and provision the ports for SCCP
- IOS CLI Enhancements
  - 4 CLIs per dial-peer = 576 CLIs for 144 ports → **2 CLIs**
  - 4 CLIs per voice-port = 576 CLIs for 144 ports → **8 CLIs**
  - **1 CLI** can configure **all the ports** in the system for SCCP
  - CLI to check if a phone is connected to a port
- Cisco Management Applications Support
  - Cisco Unified Provisioning Manager 9.0
  - Cisco Unified Operations Manager 9.0
  - BATS Tool

Feature Set and Roadmap
Analog Voice Gateway

Shipping
To Ship
Not Yet Committed

Q4 CY09 — 15.0(1)M

Features:
- Distinctive ringing for SCCP
- Enhancements to FAC support – parity with VG248

Q1 CY10 — 15.1(1)T

Features:
- SIP VMWI (DC Voltage, FSK)

Q4 CY10 — 15.1(3)T

Features:
- Secure SCCP with CUCM
- Configurable call waiting tone cadence
- Support for Media Renegotiation Call flows - 3PCC
- SNR, cBarge – across analog / IP phone
- Serviceability enhancements

Q3 CY12 — 15.2(4)M

Features:
- VG350 high density FXS & OPX-Lite analog gateway
- CLI enhancements for manageability

Future

Features:
- VG224 refresh
- SIP line side enhancements

Note: Future dates and features are estimates–subject to change

CUBE Enterprise - Update

Collaboration VT

Cisco Unified Border Element
CUBE

Enabling Session Border Control (SBC) Features on Cisco Routers

SESSION CONTROL

Call Admissions Control
Ensuring QoS
Statistics and Billing
Redundancy/ Scalability

SECURITY

Encryption
Authentication
Registration
SIP Protection
Firewall Placement
Toll Fraud

INTERWORKING

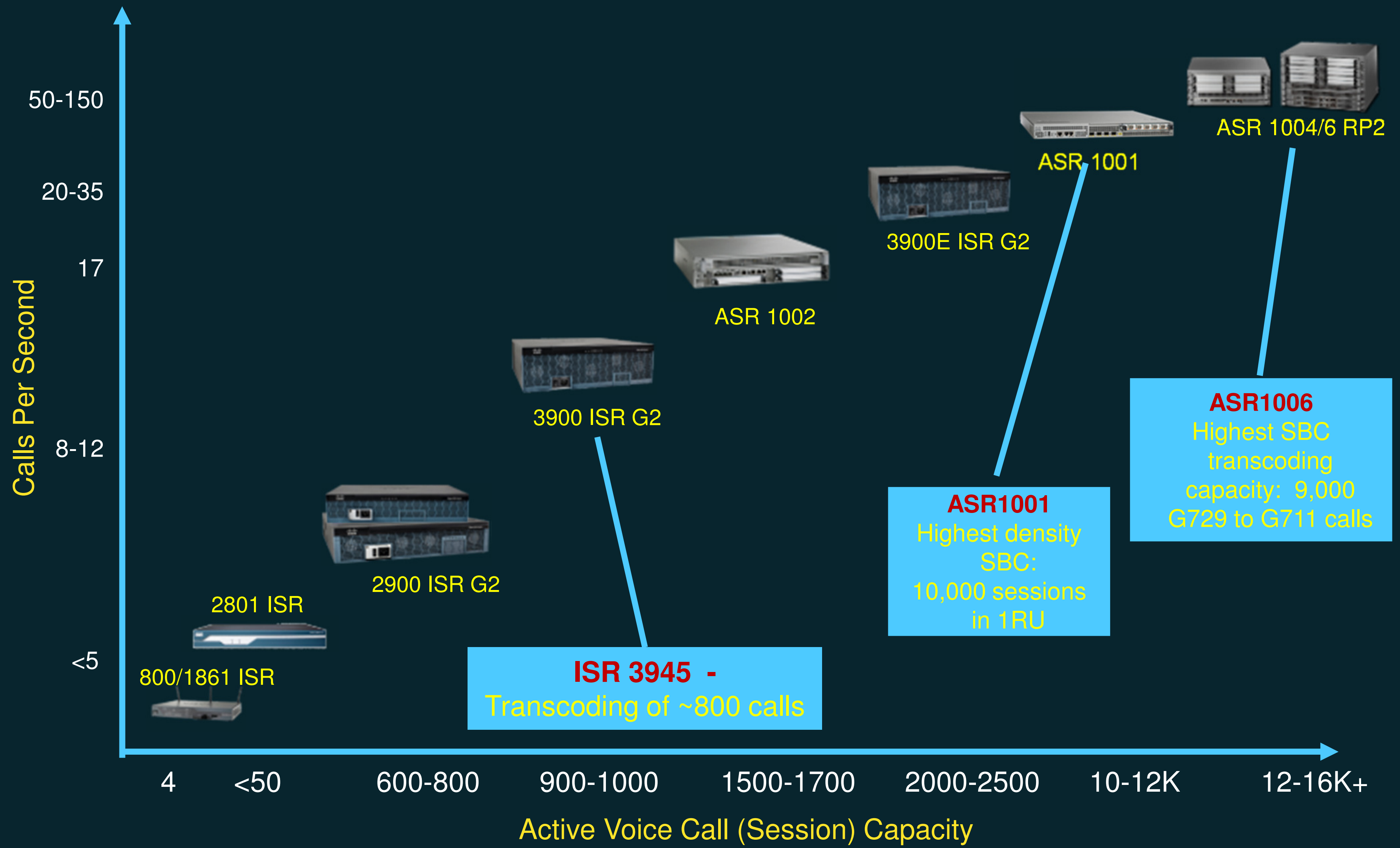
SIP - SIP
H.323 - SIP
SIP Normalization
DTMF Interworking
Transcoding
Codec Filtering

DEMARCATION

Fault Isolation
Topology Hiding
Network Borders
L5/L7 Protocol Demarcation

CUBE Scalability

Scalable Voice Trunk Capacity for Small to Large Businesses



CUBE ISR / ASR Software Releases

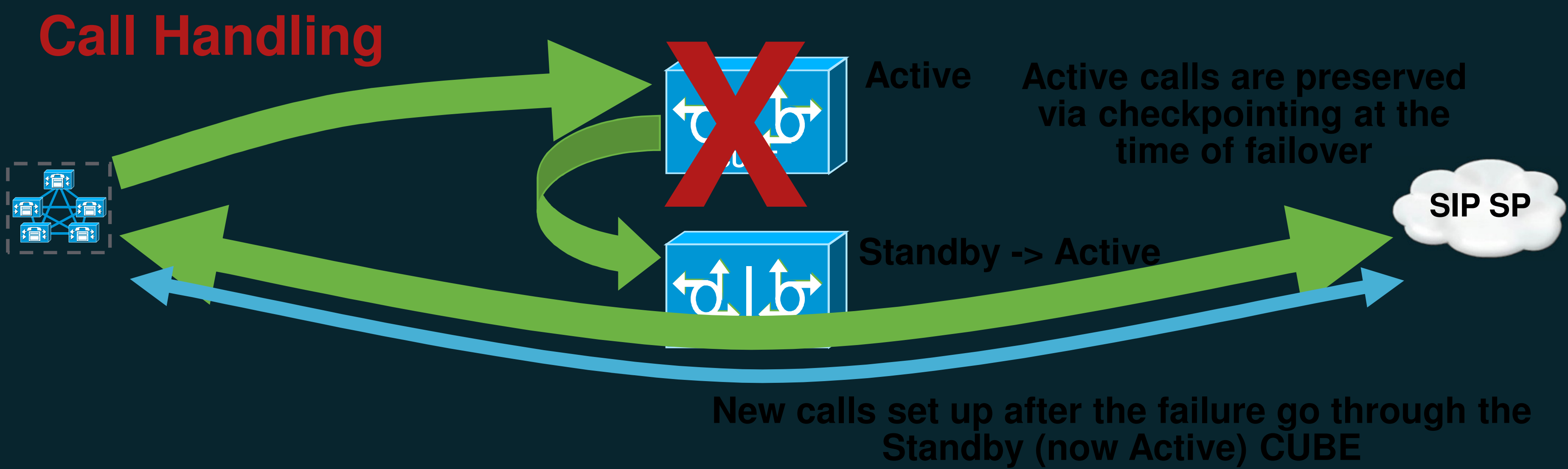
Marketable Feature Parity at Version 9.2

	ISR	ISR G2		CUBE ASR Parity with ISR	ASR		
CUBE Vers.	2800/ 3800	2900/ 3900	FCS		CUBE Vers.	IOS XE Release	FCS
1.3	12.4.22YB, 15.0.1M	15.0.1M	Oct 2009	<50%	1.3	15.0(1)S3	July 2010
1.4	15.0.1XA, 15.1.1T	15.1.1T	Apr 2010	<50%	1.4	15.0(1)S4	July 2010
8.5	15.1.2T	15.1.2T	July 2010	<50%	1.4	15.1(1)S	Nov 2010
8.6	15.1.3T	15.1.3T	Nov 2010	<50%	1.4	15.1(2)S2	Mar 2011
8.7	15.1.4M	15.1.4M	Apr 2011	~50%	1.4	15.1(3)S3	July 2011
8.8	EoL	15.2.1T	July 2011	~70%	1.5	15.2(1)S1	Nov 2011
9.0	EoL	15.2.2T	Nov 2011		NA	NA	NA
9.1	EoL	15.2.3T	Mar 2011		NA	NA	NA
NA	EoL	NA	NA	>75%	1.6	15.2(2)S	Mar 2012
NA	EoL	NA	NA	>85%	1.7	15.2(4)S	July 2012
9.2	EoL	15.2.4M	Oct 2012	>90%	9.2	15.3(1)S	Nov 2012
9.3	EoL	15.3.1	Mar 2013	>90%	9.3	15.3(2)S	Mar 2013

Internal: http://zed.cisco.com/confluence/display/VOICE/IOS+SBC+Release+Information
Cisco.com: Table 2 in the CUBE FAQ at www.cisco.com/go/cube

CUBE Stateful Failover support

- 15.1.2T added support for media preservation only
- This release adds support for signaling preservation – signaling based disconnect and other supplementary services will work post-switchover
- Also added stateful failover for Software MTP based calls on router



CUBE ISR G2 High Availability (HA) Using HSRP Configuration Example
http://www.cisco.com/en/US/products/sw/voicesw/ps5640/products_configuration_example09186a0080b40d82.shtml

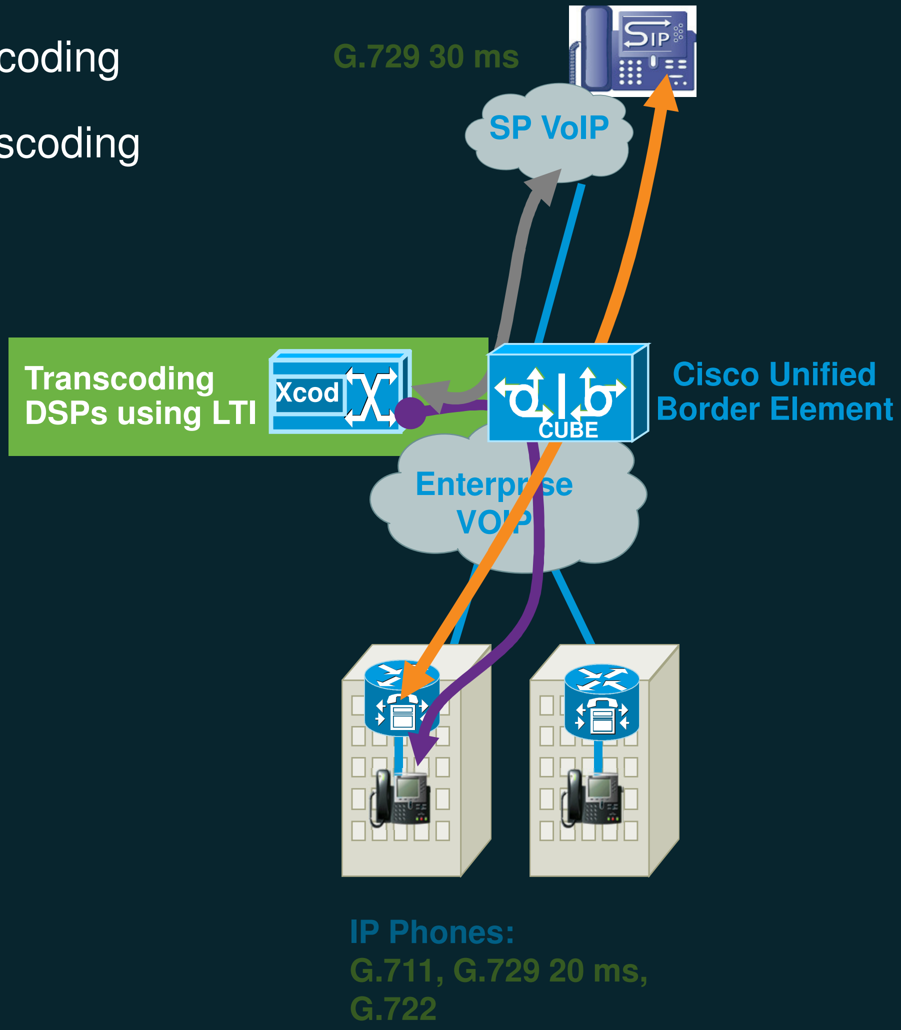
Local Transcoding Interface

- New CLI mechanism for control of DSP for transcoding
- Supported only with CUBE invoked DSP for transcoding
- LTI Supported on both ISR and ASR platform

  DSPs must be local to CUBE

- BENEFIT:
  - Simplified CLI configuration for transcoding

```
dspfarm profile 2 transcode universal
        codec g711ulaw
         codec g711alaw
        codec g729ar8
        codec g729abr8
        maximum sessions 10
        associate application CUBE
```



Configurable RTP port range

- The Per Interface Port Management feature provides the capability where the port range is managed per IP address range.
- CUBE can be configured with multiple IP address ranges, where each range shares the same port ranges with a maximum port range of 16K to 32K.

BENEFITS:

- INTEROPERABILITY - some third party applications use specific RTP port ranges. This feature allows CUBE to conform to those applications.
- SECURITY CONFIGURATION - Firewall can be given more precise range of ports that will be used for RTP traffic.

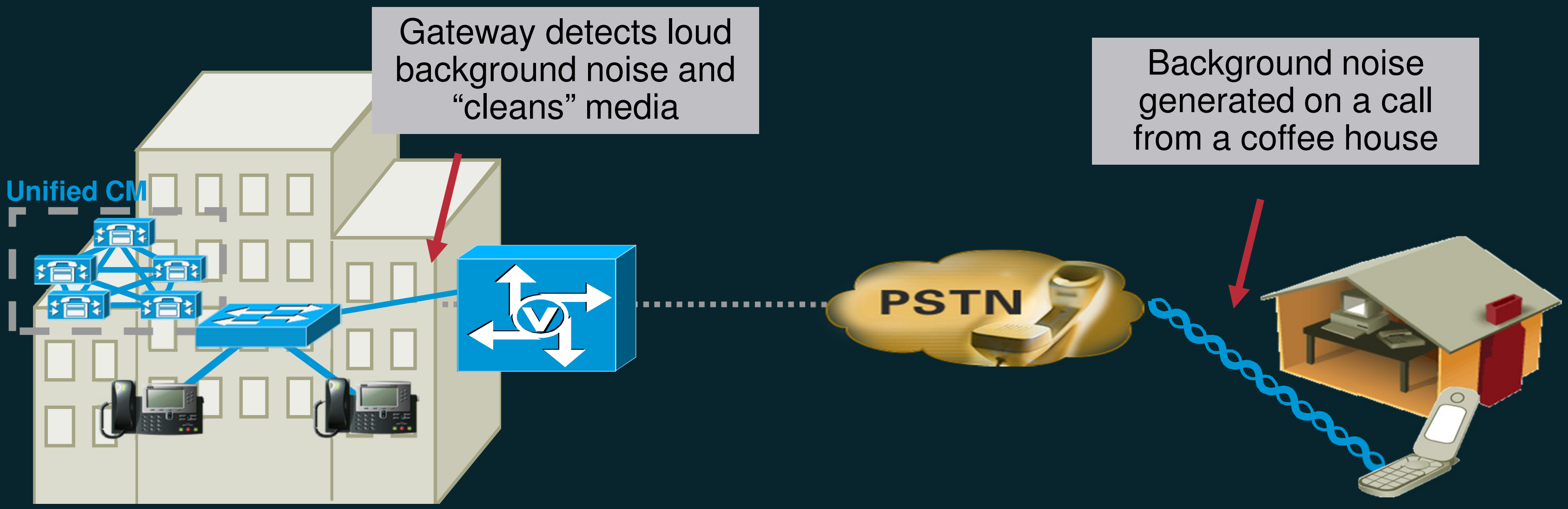
Example

```
voice service voip
  media-address range 8.8.8.1 8.8.8.2
  media-address range 9.42.30.45 9.42.30.46
  rtp-port range 20000 - 30000
```

IP Address Range	RTP Port Range
8.8.8.1 – 8.8.8.2	20000-30000
9.42.30.45 – 9.42.30.46	20000-30000
Global default (all other IP addresses)	20000-30000

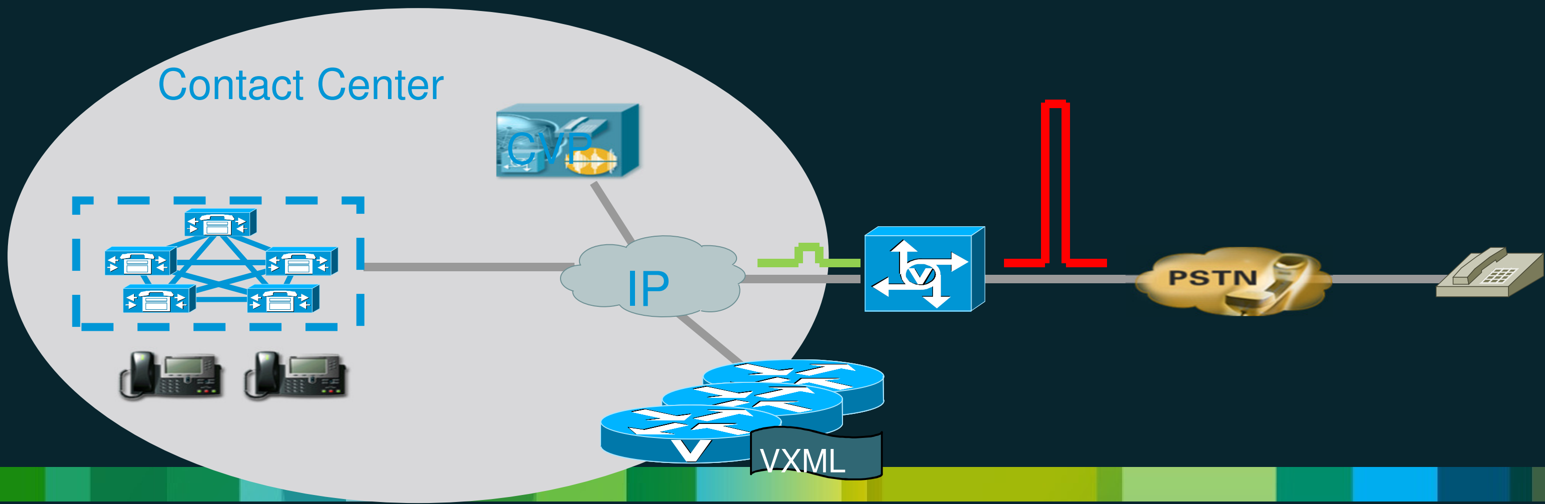
Noise Reduction

- PVDM3 used to clean audio by removing loud and distracting background noise
- Enhances user experience by delivering quality audio
- Supported on TDM-IP, CUBE and Conference calls
- Enabled globally on router or selectively per dial peer basis



Acoustic Shock Protection

- Sudden loud high pitch feedback tone, a.k.a acoustic shock can a workplace safety concern especially for call center workers.
- PVDM3 DSPs have the ability to detect acoustic shock neutralize its effect
- Detects the loud, persistent, dominant single frequency tone in a voice channel and performs corrective action
- Supported on TDM-IP, CUBE and Conference calls

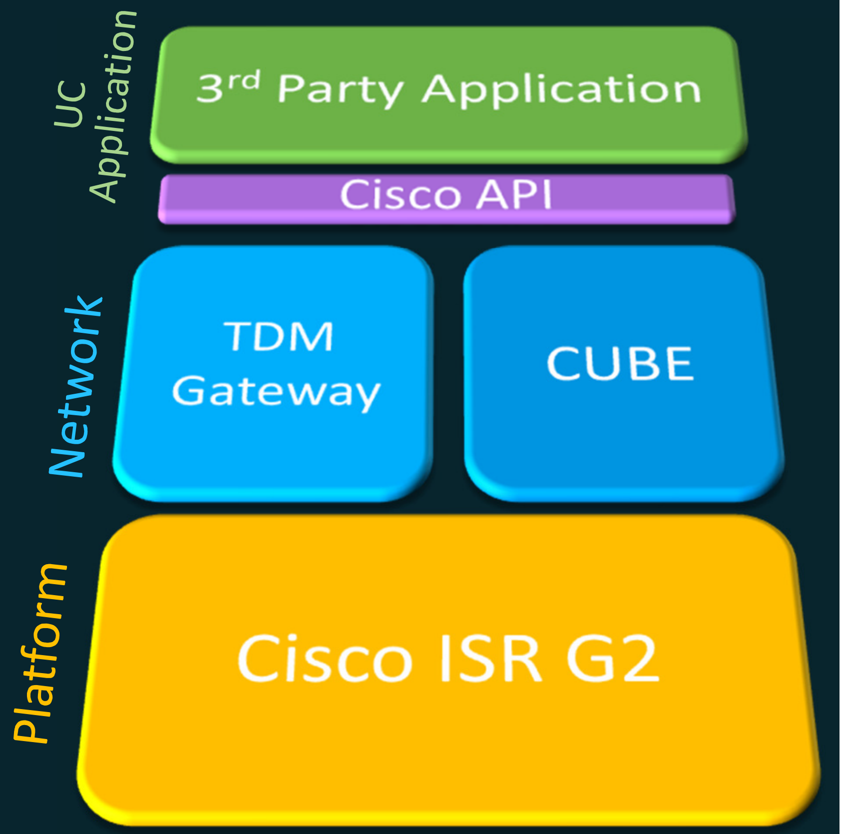


Gateway Services API

Cisco UC Gateway Services API
Monitor and Control the Voice Network Edge

API Designed to:

- Allow users or policies to dynamically control the gateways to the voice network
- Provide analytics of the voice network through real time monitoring.
- Integrate into any web application using WSDL-compatible development tools
- Allow 3rd parties to integrate applications onto Cisco ISR using UCS express
- Simplify management and architecture
    - Single platform using Cisco ISR
    - Enterprise-wide solutions (TDM+SIP)
    - Combine with data solutions



Enable Security, Visibility & Control with Voice Policy

SecureLogix
We see your voice™

Recognizing Voice Network Usage as a Strategic Business Issue

Harassing / Threatening Callers

Avoid productivity loss & safety issues

Toll Fraud

Corporations lack real-time defense

Voice Service Abuse & Theft

Ensure employee voice network use matches business objectives

Contact Center Fraud & ID Theft

Legal risk and financial losses for corporations and customers

Capacity Monitoring

Enables better network planning and staffing requirements

Unauthorized Fax or Modem Usage

Most commonly found issue

Other Developers & Examples

- NICE:
  - Using open APIs to dynamically control media recording
  - Eliminates need to record all voice calls: User-controlled or policy-controlled
  - Ideal for call centers
- MediaSense
  - Using open APIs to dynamically control media
- CUCM:
  - Enable IP-PBX to gain increased control of Gateway
- E-Loyalty / Teletech & Vodafone
  - Mobility applications
- WEBEX (Roadmap features)
  - Integrating with WebEx so end-users can turn on/off background noise reduction during conference calls

CME / SRST / CUE / E-SRST 9.0 Update

CME/SRST 9.1

Cisco Unified CME/SRST 9.1

FCS July 2012 (IOS 15.2(4)M)

NEW

KEM Module Support

Voice hunt group for E-SRST

Cisco Unified Communications Manager Express 9.1

CME/SRST now Supported on low end 88x

Support for small 5 user Branches

- Maximum flexibility in minimum footprint

Secure voice router for Managed Service Providers

- Small Site Survivability for the Cloud
- Quick Time to market

Three new platforms

- c881-V-K9, c887VA-V-K9, c887-VA-V-W-E-K9

Cisco Unity Express 8.6.3

- FCS: Q3 CY 2012
- Trivial mailbox PIN checks for end-users
- Unified Communications 9.0 inter-op
  - CUCM, CUC, CME, SRST 9.0
- 69xx & 89xx phone support for VoiceView

9.0: Cloud Telephony Survivability is here

HCS now includes Survivability!

New Features

- Support for Cisco Hosted Collaboration Solution (HCS)

Customer Benefits

•Ensures telephony reliability for cloud delivery

•Maintain business continuity during outages

Partner Benefits

•Easier sales process – helps overcome customer concerns about cloud reliability

Enhanced SRST solution Cisco Unified SRST Manager 9.0

Cisco Has Branch Telephony Survivability Options to Meet All Needs

SRST

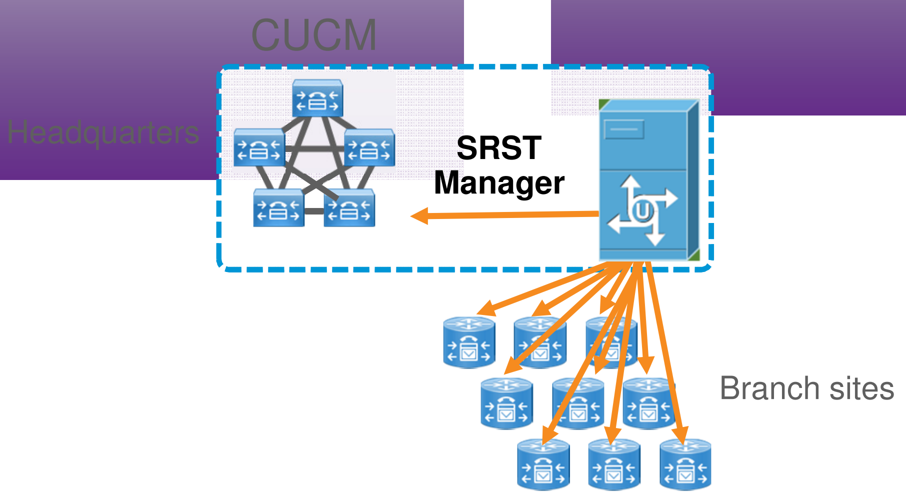
Ideal for customers who require:

- Basic phone features during failover
- Support of SIP & SCCP phones
- Supports Cisco Unified CM and Cisco Unified CMBE

E-SRST

Ideal for customers who require:

- Full phone features during failover
- Hunt Group, Park Pickup, Speedials
- Simplified administration
- Automated provisioning & deployment
- Better security
- Supports Cisco Unified CM only



Cisco Unified SRST Manager 9.0

FCS Aug 2012

NEW

Virtualization

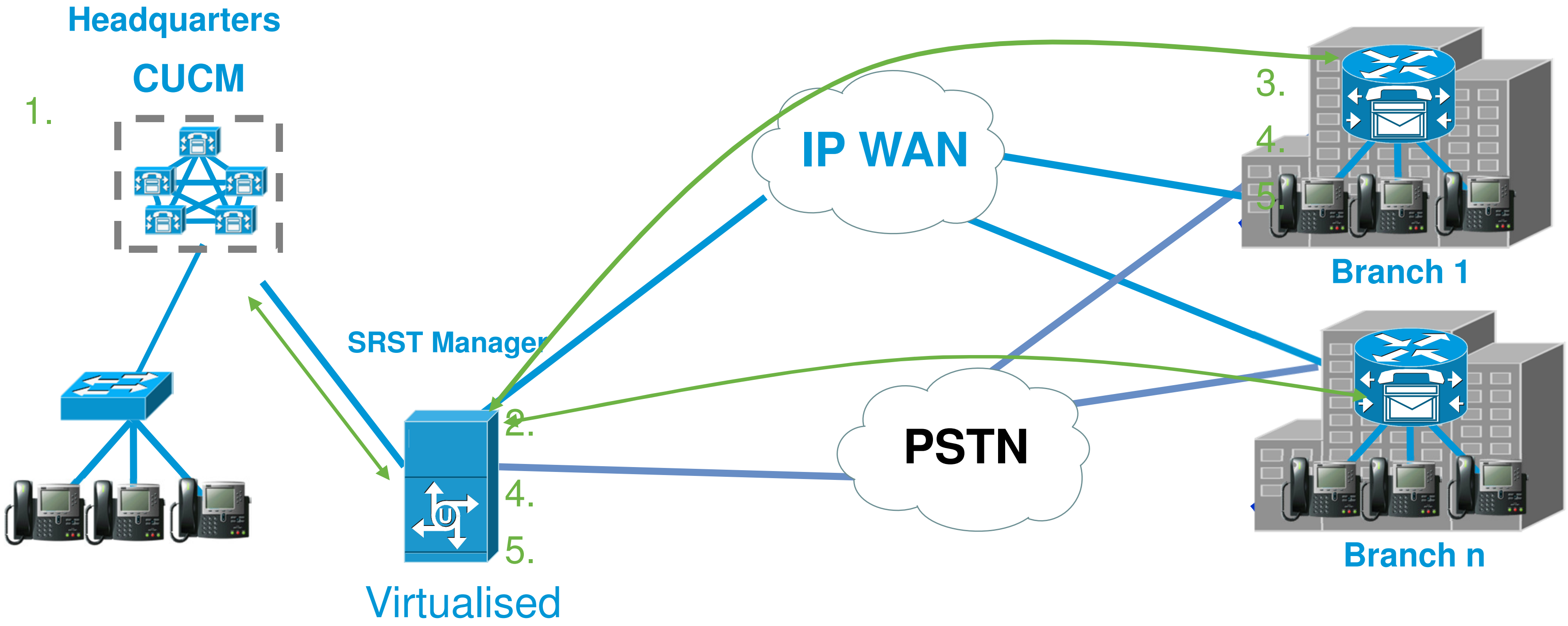
SIP Phones

Classic SRST

Dialplan Management

Enriching the Survivability Experience

E-SRST– Solution Provisioning



1. Provision the branch in CUCM
2. Configure SRST Manager to connect to CUCM for provisioning info
3. Provision branch router for IP connectivity and credentials
4. SRST Manager sends provisioning information to each branch Router for Phone users , dialplan restrictions etc
5. SRST Manager maintains any moves, adds and changes across HQ and branches

New CCO Websites

Worldwide [change] | Log In | Account | Register | My Cisco

Products & Services | Support | How to Buy | Training & Events | Partners

Voice and Unified Communications

http://www.cisco.com/go/ucgatewaysandservices

Cisco Unified Communications Gateways and Services

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Save, Simplify, Extend

Use Cisco Integrated Services Routers to ease transition to SIP trunking and more. (3:39 min)

Featured Products | All Products | Learn More

Unified Communications Gateways and Services

Cisco offers flexible platforms with unified communications services for all types of gateway and session border control deployments.

Cisco Integrated Services Routers (ISRs)

- Deliver modular support for a multitude of unified communications services
- Ideal for teleworkers, small and medium businesses, or enterprise branch offices
- Deploy multiple services on the same platform in the same footprint
- Services include gateway, session border control, survivability, and more

Learn More

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Empower Your Branch Offices

Make your branch offices as productive as your headquarters location.

Read White Paper

Transform Communications

Integrate SIP trunks in your enterprise network for unified communications.

Read White Paper

Voice and Unified Communications

http://www.cisco.com/go/uconisr

Unified Communications on Cisco Integrated Services Routers

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VOICE AND UNIFIED COMMUNICATIONS

Unified Communications on Cisco Integrated Services Routers

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Cisco Integrated Services Router (ISR) Platforms

Deliver modularity for unified communication services to teleworkers, small and medium-sized businesses, or branch offices.

Cisco 3900 Series ISRs

- Offers a high-performance gateway and session border control platform
- Ideal for mid-size to large deployments
- Modular setup offers the ability to host additional unified communications services

Learn More

Cisco 2900 Series ISRs

- Offers a high-performance gateway and session border control platform
- Ideal for small to mid-size deployments
- Modular setup offers the ability to host additional unified communications services

Learn More

Cisco 1861E ISR

- Designed for small offices that need a gateway, Cisco Unified Communications Manager Express (CME), or Cisco Unified

Learn More

Let Us Help

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US/CAN | 5am-5pm Pacific
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Save, Simplify, Extend

Ease your network evolution to SIP trunking and beyond.

Watch Video

SIP Trunking Deployment Models White Paper

Learn the key factors to consider when evaluating and designing a SIP trunk network, including architecture and migration options.

Learn More

Thank you.